Reg No :______________ Name:__________________________

APJ ABDUL KALAM TECHNOLOGICAL UNIVERSITY

FIFTH SEMESTER B.TECH DEGREE EXAMINATION, DECEMBER 2018

Course Code: EC301

Course Name: DIGITAL SIGNAL PROCESSING

Max. Marks: 100 Duration: 3 Hours

PART A

Answer any two full questions, each carries 15 marks. Marks

1 a) Given x(n) = { 1, -2, 3, -4, 5, -6 } without calculating DFT find the following quantities? (5)

a) X(0) b) $\sum_{K=0}^{5} X(K)$ c) X(3) d) $\sum_{K=0}^{5} |X(K)|^2$ e) $\sum_{K=0}^{5} -1^K X(K)$

b) Find the convolution of x(n) = {1, 2, 3, 4, 5} and h(n) = {1, 1, 1} using overlap save method? (5)

c) State Circular frequency shift property of DFT? (5)

4 –point DFT of the signal x(n) ={a, b, c, d} is X(K). Find the IDFT of X(K-2) ?

2 a) Find the number of complex multiplications and additions involved in the calculation of 1024 DFT using direct computation and radix2 FFT algorithm? (4)

b) How will you obtain linear convolution from circular convolution? For x(n) ={1, 2, 3} and h(n) = {-1, -2}, obtain linear convolution x(n)*h(n) using circular convolution? (5)

c) Given at g(n) ={1, 0, 1, 0} and h(n) = {1, 2, 2, 1} find the 4 point DFTs of these sequences using a single 4 point DFT. (6)

3 a) Describe the steps involved in radix 2 DIT FFT algorithm (5)

b) Find the DFT of the sequence {1, 2, 3, 4, 4, 3, 2, 1} using DIT algorithm (7)

c) What do you mean by in place computation of DFT? (3)

PART B

Answer any two full questions, each carries 15 marks.

4 a) Explain the significance of linear phase FIR filter and comment on its impulse response? (4)

b) Design an ideal lowpass filter with frequency response (6)

$H(e^{j\omega}) = 1$ for $-0.5\pi \le \omega \le 0.5\pi$ and $H(e^{j\omega}) = 0$ for $0.5\pi \le |\omega| \le \pi$.

Find h(n) for N = 11.(use rectangular window)

c) Determine the frequency response of FIR filter defined by (5)

y(n) = 0.25x(n) + x(n - 1) + 0.25x(n - 2). Calculate the phase delay and group delay?

5 a) Convert the analog filter *H(s)* given below in to a second order Butterworth digital filter using impulse invariance technique. (6)

$$H(s) = \frac{1}{s^2 + \sqrt{2}\,s + 1}$$

b) Why can't we use impulse invariance technique for implementing digital highpass filter? (4)

c) Describe the steps involved in the design of digital Butterworth bandpass filter? (5)

6 a) Derive the equation for cutoff frequency in Butterworth filter? (5)

b) Apply bilinear transformation to $H(s) = \frac{2}{(s+1)(s+2)}$ with T = 1 sec and find H(z)? (5)

c) What is warping effect in bilinear transformation method and how can we eliminate it? (5)

PART C

Answer any two full questions, each carries 20 marks.

7 a) Draw the block diagram of TMS320C67XX and explain functions of each block? (10)

b) Realize the system function using minimum number of multipliers (5)

$H(z) = (1 + z^{-1})\,(1 + 0.5z^{-1} + 0.5z^{-2} + z^{-3})$

c) Obtain the transposed directform II structure for the system (5)

$y(n) = 0.5y(n - 1) - 0.25y(n - 2) + x(n) + x(n - 1)$

8 a) Realize the system given by difference equation $y(n) = -0.1y(n - 1) + 0.2y(n - 2) + 3x(n) + 3.6x(n - 1) + 0.6x(n - 2)$ in cascade form? (6)

b) Obtain the parallel form realization for above system (6)

c) Find the lattice structure implementation of FIR filter h(n) = {1, 13/24, 5/8, 1/3} (8)

9 a) Explain the effect of coefficient quantization in IIR and FIR filters? (10)

b) If quantization noise has uniform distribution with zero mean, find the quantization noise in ADC with step size Δ? (5)

c) A signal x(n) is obtained by sampling analog signal x(t) at twice the Nyquist rate. If we wish to down sample x(n) by a factor 4, obtain the bandwidth of the decimation filter required for supressing aliasing distortion. (5)

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Reg No :- Name:-

APJ ABDUL KALAM TECHNOLOGICAL UNIVERSITY

FIFTH SEMESTER B.TECH DEGREE EXAMINATION(S), MAY 2019

Course Code: EC301

Course Name: DIGITAL SIGNAL PROCESSING

Max. Marks: 100 Duration: 3 Hours

PART A

Answer any two full questions, each carries 15 marks. Marks

1 a) Find the 4-DFT and 8-DFT of the sequence {1, 1, 1, 0}. Plot |X(K)| and comment on the significance of N? (10)

b) State Parseval's property? (5)
DFT of a real valued signal X(K). = {j, 1+j, A, 1-j, -1, B, -1-j, C}. Find the energy of the signal?

2 a) Find the convolution of x(n) = {1, 2, 3, 4, 5, 6, 7, 8, 9} and h(n) = {2, 4, 6} using overlap add method? (6)

b) Find the response of an LTI system with impulse response h(n) = { 1, 2, 2, 1} for an input x(n) = {1 , -1 , 1, -1 } using circular convolution? (4)

c) If x(n) = {1, 2, 3, 4}. Find DFT[DFT(x(n))] without calculating DFT? (5)

3 a) Explain the radix-2 DIT FFT algorithm and draw the corresponding flow diagram for 16 DFT computation. (10)

b) Explain about the efficient computation of DFT of a 2N- point real sequence (5)

PART B

Answer any two full questions, each carries 15 marks.

4 a) Derive equations for magnitude and phase responses of FIR filter whose impulse response is symmetric and length N odd. (5)

b) Design an ideal 6th order linear phase lowpass filter with frequency response (6)
$H(e^{j\omega}) = 1$ for $-0.5\pi \leq \omega \leq 0.5\pi$ and $H(e^{j\omega}) = 0$ for $0.5\pi \leq |\omega| \leq \pi$.
Use Hamming window.

c) Explain Gibb's phenomenon. (4)

5 a) Determine the filter coefficients of a linear phase FIR filter of length N = 15, which has a symmetric impulse response and a frequency response that satisfies the conditions, (10)

$$H\left(\frac{2\pi k}{15}\right) = \begin{cases} 1, & k = 0,1,2,3 \\ 0.4, & k = 4 \\ 0, & k = 5,6,7 \end{cases}$$

b) Prove that the zeros of FIR filter exists as reciprocals. (5)

6 Design a digital Butterworth filter that has -1dB pass band attenuation at 200 Hz and at least -15dB stop band attenuation at 540 Hz. Sampling frequency = 2000 Hz. Find the cut off frequency by matching pass band criterion. Use Bilinear transformation (T = 1 sec) (15)

PART C

Answer any two full questions, each carries 20 marks.

7 a) Explain the steps through which we obtained direct form II realization of recursive LTI system described by difference equation. (10)

$$y(n) = -\sum_{k=1}^{N} a_k \, y(n-k) + \sum_{k=0}^{M} b_k \, x(n-k)$$

b) Draw the architecture block diagram of TMS320C67XX processor (5)

c) Obtain the transposed direct form II structure for the system (5)

y(n) = 2y(n - 1) + 3y(n - 2) + x(n) + 2 x(n - 1) + 3x(n - 2)

8 a) Find the impulse response h(n) of a FIR filter, if the reflection coefficients are $K_1 = 2/5$, $K_2 = 4/21$, $K_3 = 1/8$. (6)

b) What is transposition theorem and transposed structure? (6)

c) Obtain direct form II and cascade structure for the transfer function given below. (8)

$$H(z) = \frac{1 + 2z^{-1} + z^{-2}}{1 - \frac{3}{4} z^{-1} + \frac{1}{8} z^{-2}}$$

9 a) Explain the effect of coefficient quantization in IIR and FIR filters? (10)

b) What are the main features of DSP processor? (5)

c) Explain the effect in the spectrum of a signal x(n) when it is (5)

(i) Decimated by a factor 3

(ii) Interpolated by a factor 2

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Reg No.:_______________ Name:__________________________

APJ ABDUL KALAM TECHNOLOGICAL UNIVERSITY

FIFTH SEMESTER B.TECH DEGREE EXAMINATION(R&S), DECEMBER 2019

Course Code: EC301

Course Name: DIGITAL SIGNAL PROCESSING

Max. Marks: 100 Duration: 3 Hours

PART A

Answer any two full questions, each carries 15 marks. Marks

1 a) StateParseval's theorem of DFT?Using DFT find the energy of the sequence$x(n) = 0.2^n\ u(n)\,, n < 4$. (7)

b) Compute 8-point DFT of the sequence x(n)= $\{\frac{1}{2},\frac{1}{2},\frac{1}{2},\frac{1}{2},0,0,0,0\}$ using DITFFT algorithm. Follow exactly the corresponding flow graphs and keep track of all the intermediate quantities by putting them on diagram. (8)

2 a) Find linear convolution of the sequences x(n) and z(n) using circular convolution. Given x(n)={1,2,3,1} and z(n)={4,3,2}. (7)

b) Explain how N point DFTs of two real sequences can be found using by computing a single DFT. Illustrate with the sequences x_1(n)={4,3,-1,5} and x_2(n)={6,-4,2,5}. (8)

3 a) Find the number of real multiplications and additions involved in the computation of 64-point DFT using i) direct computation ii) FFT algorithm.Also comment on the computational advantage of FFT algorithm over the direct method. (7)

b) Using Overlap Add method, find the output of the filter with filter response h(n) when an input x(n)={1,2,2,3,4,2,2,1,1} is given. Take data block size of length L= 3 and h(n)= {2,3,4}. (8)

PART B

Answer any two full questions, each carries 15 marks.

4 a) Design a linear phase FIR low pass filter with cut off frequency of 2 kHz and sampling rate of 8 kHz with a filter length 11 using Hanning window. (10)

b) Find the filter transfer function H(z) from the analog filter with system function H(s) using Impulse Invariance method. (5)

$$H(s) = \frac{s+1}{s^2 + 0.2s + 16.01}$$

5 a) Apply frequency sampling technique to design a linear phase FIR filter of length N=7 with following specification. (10)

$$H_d(e^{j\omega}) \quad = e^{-j\alpha\omega}; \quad 0 \le |\omega| \le 0.55\pi$$
$$= 0 \quad otherwise$$

b) Transform the prototype low pass filter with system function $H(s) = \frac{\Omega c}{s+\Omega c}$ into high pass and band pass filters. (5)

6 a) Design a Butterworth low pass digital IIR filter witha pass band edge frequency of**0.25π** with a ripple not exceeding **0.5 dB** and a minimum stop band attenuation **15dB** with a stop band edge frequency of **0.55π**. Use bilinear transformation. (10)

b) Compare the performance of FIR filter design using rectangular window and Hamming window. (5)

PART C

Answer any two full questions, each carries 20 marks.

7 a) Determine a direct form realization of the FIR filter with the following filter function using minimum number of multipliers. (4)

h(n)={1,2,3,4,3,2,1}

b) Draw the cascade and parallel form realisation of the filter with following transfer function (8)

$$H(z) = \frac{3(5-2z^{-1})}{\left(1+\frac{1}{2}z^{-1}\right)(3-\ z^{-1})}$$

c) How upsampling and downsampling by a factor of 3 affect the frequency spectrum of a signal x(n) with frequency spectrum $X(e^{j\omega})$? What is the need of low pass filter prior to downsampling? (8)

8 a) For the signal $x(n) = 0.2^n\ u(n), n \le 8$, plot the following signals (4)

(i) x(n) downsampled by 3 (ii) x(n) upsampled by 3

b) With an example illustrate the error introduced by truncation and rounding in fixed point representation of numbers. (8)

c) What is the effect of coefficient quantization in IIR filter structures? (8)

9 a) Obtain the direct form II, cascade and transposed direct form II structures for the (10)

system.

$$y(n) = -0.1y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$$

b) Explain the architecture of TMS320C67xx DSP with block diagram. (10)

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Reg No.:_______________ 00000EC301121901 Name:___________________________

APJ ABDUL KALAM TECHNOLOGICAL UNIVERSITY

Fifth semester B.Tech degree examinations (S) September 2020

Course Code: EC301

Course Name: DIGITAL SIGNAL PROCESSING

Max. Marks: 100 Duration: 3 Hours

PART A

Answer any two full questions, each carries 15 marks. Marks

1	a)	Compute 5 point DFT of the sequence $x(n) = \{1,1,1,1,1\}$	(5)
	b)	Express DFT as a linear transformation. How many complex multiplications and additions are needed to compute N point DFT.	(10)
2	a)	Find the 4 point circular convolution of sequences $x_1(n) = \{2,1,2,1\}$ with $x_2(n) = \{1,2,3,4\}$	(8)
	b)	Explain how to compute linear convolution of two sequences of length N_1 and N_2 using DFT.	(7)
3	a)	Derive Decimation In Time (DIT) FFT algorithm for 8 point DFT and draw the signal flow graph.	(8)
	b)	Explain overlap and add method for filtering of long data sequences.	(4)
	c)	Prove that N point DFT is periodic with period N	(3)

PART B

Answer any two full questions, each carries 15 marks.

4	a)	How the phase of a filter is related to frequency for a linear phase filter? Why linear phase is important in certain filtering applications?	(5)
	b)	Derive the condition for impulse response $h(n)$ for getting a linear phase response. Assume length of $h(n) = N$, an even number.	(10)
5	a)	Derive the mapping between s and z used in bilinear transformation.	(3)
	b)	Design a digital Butterworth filter satisfying the constraints $$0.6 \leq \left\|H\left(e^{j\omega}\right)\right\| \leq 1;\ 0 \leq \omega \leq 0.35\pi$$ $\left\|H(e^{j\omega})\right\| \leq 0.1; 0.7\pi \leq \omega \leq \pi$. Use Bilinear transformation. Assume $T = 0.1$	(12)
6	a)	Give equations for N point Hamming and Hanning Window functions. Compare them in terms of main lobe width and side lobe level.	(6)
	b)	Explain frequency sampling method of FIR filter design.	(3)

c) $$\text{Let } H_d(\omega) = e^{-j3\omega}; 0 \le |\omega| \le \frac{\pi}{2};$$ (6)

$$= 0; \frac{\pi}{2} \le \omega \le \pi$$

Get the filter coefficients for FIR filter using frequency sampling. Assume *N*= 7.

PART C

Answer any two full questions, each carries 20 marks.

7 a) Draw the direct form 1 and direct form 2 structures for the difference equation $y(n) = x(n) + 0.5x(n-1) + 3y(n-1) - 2y(n-2)$. (10)

b) Draw the block diagram of TMS320C67xx and briefly explain function of all blocks. (10)

8 a) Explain the effects of coefficient quantization in FIR and IIR filters. (10)

b) Derive the variance of quantization noise in ADC. Assume step size is Δ. (5)

c) Let $x(n) = 0.5^{\,n}u(n)$. Obtain the signals for decimation by 3, interpolation by 3. (5)

9 a) Find the lattice structure implementation of the FIR filter with $h(n) = 1,0.5,0.75,-0.6$ (10)

b) Write notes on finite word length effects in DSP systems. (5)

c) Let a signal $x(n) = 0.5^{\,n}u(n)$ is decimated by 2. What happens to its spectrum? (5)

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